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SERIES P: TELEPHONE TRANSMISSION QUALITY, TELEPHONE INSTALLATIONS, LOCAL LINE NETWORKS

Subscribers' lines and sets

Transmission performance of group audio terminals (GATs)

ITU-T Recommendation P.300

ITU-T P-SERIES RECOMMENDATIONS

TELEPHONE TRANSMISSION QUALITY, TELEPHONE INSTALLATIONS, LOCAL LINE NETWORKS

Vocabulary and effects of transmission parameters on customer opinion of transmission quality	Series	P.10
Subscribers' lines and sets	**Series**	**P.30** **P.300**
Transmission standards	Series	P.40
Objective measuring apparatus	Series	P.50 P.500
Objective electro-acoustical measurements	Series	P.60
Measurements related to speech loudness	Series	P.70
Methods for objective and subjective assessment of quality	Series	P.80 P.800
Audiovisual quality in multimedia services	Series	P.900

For further details, please refer to the list of ITU-T Recommendations.

ITU-T Recommendation P.300

Transmission performance of group audio teminals (GATs)

Summary

This Recommendation provides the speech transmission requirements for Group Audio Terminals (GATs). The associated test methods for verifying the specified performances are contained in Annex A. An *in-situ* procedure for checking and aligning GAT terminals in the field is provided in Appendix I.

Reference is made, as far as possible, to other Recommendations applicable to speech terminals for analogue and digital networks. Only terminal configurations not already covered by these Recommendations are addressed in detail.

Source

ITU-T Recommendation P.300 is a revision of ITU-T Rec. P.30 (1998). It was revised by ITU-T Study Group 12 (2001-2004) and approved under the WTSA Resolution 1 procedure on 29 November 2001.

FOREWORD

The International Telecommunication Union (ITU) is the United Nations specialized agency in the field of telecommunications. The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of ITU. ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The World Telecommunication Standardization Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 1.

In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

NOTE

In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

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As of the date of approval of this Recommendation, ITU had not received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementors are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database.

CONTENTS

ITU-T Recommendation P.300

Transmission performance of group audio terminals (GATs)

1 Scope

This Recommendation provides the audio performance requirements and test methods for Group Audio Terminals (GATs), i.e. terminals which are specifically designed for being used by several users at the same time.

GATs cover a wide range of products, ranging from the hands-free telephone set, when used by several users at the same time, to the teleconference audio terminals incorporating sophisticated echo control mechanisms. GATs can be designed to operate both on the analog POTS networks and on digital ISDN links. While the former can only offer telephone bandwidth performances, the latter can be designed to offer either narrow-band or wideband audio communication facilities.

This Recommendation addresses this whole range of devices by identifying, as far as possible, the already existing Recommendations applicable to each terminal type.

2 Normative references

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision: all users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published.

[1] ITU-T Recommendation P.310 (2000), *Transmission characteristics for telephone-band (300-3400 Hz) digital telephones*.

[2] ITU-T Recommendation P.311 (1998), *Transmission characteristics for wideband (150-7000 Hz) digital handset telephones*.

[3] ITU-T Recommendation P.57 (1996), *Artificial ears*.

[4] ITU-T Recommendation P.10 (1998), *Vocabulary of terms on telephone transmission quality and telephone sets*.

[5] ITU-T Recommendation P.340 (2000), *Transmission characteristics and speech quality parameters of hands-free telephones*.

[6] ITU-T Recommendation P.342 (2000), *Transmission characteristics for telephone band (300-3400 Hz) digital loudspeaking and hands-free telephony terminals*.

[7] ITU-T Recommendation P.341 (1998), *Transmission characteristics for wideband (150-7000 Hz) digital hands-free telephony terminals*.

[8] ITU-T P-Series Recommendations – Supplement 16 (1988), *Guidelines for placement of microphones and loudspeakers in telephone conference rooms [1] and for Group Audio Terminals (GATs)*.

[9] ITU-T Recommendation P.51 (1996), *Artificial mouth*.

[10] ITU-T Recommendation P.79 (1999), *Calculation of loudness ratings for telephone sets*.

[11] ISO 3:1973, *Preferred numbers – Series of preferred numbers*.

[12] ITU-T Recommendation G.122 (1993), *Influence of national systems on stability and talker echo in international connections*.

[13] IEC 60651 (2001), *Sound level meters*.

[14] ITU-T Recommendation P.50 (1999), *Artificial voices*.

3 Definitions and abbreviations

Relevant definitions and abbreviations in ITU-T Rec. P.10 [4] apply.

For the purposes of this Recommendation, the following terms are additionally defined:

3.1 Group Audio Terminal (GAT): Terminal designed for being used by several users at the same time and specifically intended for accessing point-to-point and multipoint teleconference sessions.

3.2 reference codec: Practical implementation of an ideal codec. Used for carrying out speech transmission measurements on digital terminal equipment by means of analogue instrumentation. The Reference Codec shall provide a 0 dBr analogue interface to the testing instrumentation and implement the relevant coding algorithms (e.g. ITU-T Rec. G.711 for telephone band measurements and ITU-T Rec. G.722 for wide band applications).

3.3 reference sphere: One meter radius sphere inside an echo-free environment, where anechoic conditions exist within the tolerance limits specified in ITU-T Rec. P.341 [7] (Table A.1/P.341). Three orthogonal axes are defined through the center of the Reference Sphere: x and y (horizontal) and z (vertical).

The following abbreviations are used:

d_r Loudspeaker-listener nominal operating distance of hands-free multiple users GATs

d_s Microphone-speaker operating distance of hands-free multiple users GATs

F_r Correction factor for receiving hands-free measurements of hands-free multiple users GATs

F_s Correction factor for sending hands-free measurements of hands-free multiple users GATs

F_{tcl} Correction factor for terminal coupling loss measurements of hands-free multiple users GATs

ISDN Integrated Services Digital Network

POTS Plain Old Telephone Service

RLR Receiving Loudness Rating

SLR Sending Loudness Rating

4 Group Audio Terminals types

The GAT terminals are intended for supporting communications between groups of users, either point-to-point or multipoint.

The following types of audio facilities can be provided by GAT terminals, either separately or in combination:

– handsets;

– headsets;

– hands-free (single-user facilities);

– hands-free (multiple-users facility).

GATs can be designed to provide telephone-band and/or wideband audio facilities.

5 Speech transmission characteristics

The GAT terminals shall be able to interwork (end-to-end) with analogue (POTS) and digital (ISDN) narrow-band telephone sets, as well as with ISDN videophones and wideband telephones. As a consequence, the audio standards applicable to these devices also apply to GAT terminals intended for the same network access, as appropriate and as specified below.

5.1 Handset mode

For GATs designed for interconnecting with the analogue network access, the same relevant regional requirements applicable to POTS telephone sets are also applicable.

For GATs designed for interconnecting with the ISDN network access, the requirements and measurement methods of ITU-T Recs. P.310 [1] and P.311 [2] apply, respectively for telephone-band and wideband operations.

5.2 Headset mode

In principle, the requirements for headset operations shall be based on the requirements for handset operations, with the following allowances:

- the suitable positioning of the microphone with respect to the Mouth Reference Point (MRP) position shall be stated by the manufacturer;
- an appropriate Artificial Ear shall be used and the measurement results shall be referred to the Ear Reference Point (ERP) position as specified in ITU-T Rec. P.57 [3].

5.3 Hands-free mode (single-user facilities)

The following Recommendations apply to GATs terminals designed for providing hands-free facilities to multiple users by combining more single-user facilities.

ITU-T Rec. P.340 [5] applies to GATs designed for the analogue network access.

The requirements and measurement methods of ITU-T Recs. P.342 [6] and P.341 [7] respectively apply for GATs intended for connection to the ISDN and implementing telephone-band (ITU-T Rec. G.711) or wide band (ITU-T Rec. G.722) speech coding.

5.4 Hands-free mode (multiple-users facility)

Multiple users GAT terminals are typically designed for guaranteeing optimum hands-free audio performance when installed in "typical" conference rooms (i.e. in "live" environments with acoustic reverberation phenomena and background noise). To this purpose, suitable signal processing techniques are frequently used for eliminating the reverberation effects in the transduced signals and/or suppressing the room noise. Performance under real operating conditions can then only be evaluated in the actual installation environments, whose characteristics can not be specified in a general way.

This Recommendation is not then intended to address all the installation conditions of GAT terminals, but specifies their audio performances under reference echo-free conditions. In case this testing environment is not compatible with the technical characteristics of specific terminals, their manufacturers are given the option to state alternative, more suitable, acoustic testing environments for carrying out their characterisation.

An *in-situ* testing procedure for checking the correct audio alignment of installed terminals is provided in Appendix I.

Guidelines for the placement of microphones and loudspeakers in conference rooms can be found in Supplement 16 of the P-series Recommendations [8].

5.4.1 Receiving volume control and sensitivity adjustments

Due to the wide range of operating conditions, both the sending and receiving sensitivity of multiple-user handsfree terminals are generally adjusted at their installation on the basis of the room characteristics and of the distance between the transducers and the users. These adjustment controls shall not be accessible to normal users. An additional user-accessible receiving volume control (referred to in the following as "receiving volume control") can be provided in order to allow the user to establish the optimum listening conditions on the basis of the actual room noise, the number of participants and of other variable factors.

5.4.1.1 Sending sensitivity adjustment

The sending sensitivity adjustment is intended to allow for varying the positioning of the microphone(s) with respect to the user(s), according to the installation environment constraints. The manufacturer shall state the regulation range of this control and the microphone-to-speaker nominal operating distance ($\mathbf{d_s}$) for which the terminal submitted to the tests has been adjusted.

5.4.1.2 Receiving sensitivity adjustment

The receiving sensitivity adjustment is intended to allow for varying the positioning of the loudspeaker(s) with respect to the user(s) according to the installation environment constraints. The manufacturer shall state the regulation range of this control and the loudspeaker-to-listener nominal operating distance ($\mathbf{d_r}$) for which the terminal submitted to the tests has been regulated.

5.4.1.3 Receiving volume control

Unless stated otherwise, the compliance requirements here provided refer to the maximum position (maximum sensitivity) of the receiving volume control (when manually adjustable).

The dynamic range of the receiving volume control (when manually adjustable) shall be of at least 15 dB.

5.4.1.4 Adaptive gain control

An adaptive gain control, sensible to the background noise level, may be implemented into the GAT. Its action should result into complementary and symmetrical gain variations, in the receiving and in the sending paths, as a function of the ambient noise level.

5.4.2 Sensitivity-frequency responses

ITU-T Rec. P.340 [5] applies to GATs designed for the analogue network access.

The requirements of ITU-T Recs. P.342 [6] and P.341 [7] respectively apply for GATs intended for connection to the ISDN and implementing telephone-band (ITU-T Rec.G.711) or wideband (ITU-T Rec. G.722) speech coding.

5.4.3 Loudness Rating

ITU-T Rec. P.340 [5] applies to GATs designed for the analogue network access.

For GATs intended for connection to the ISDN the following applies.

5.4.3.1 Sending

The nominal value of Sending Loudness Rating (SLR) shall be:

$$SLR = (+\,13 – F_s)\ \text{dB}$$

where F_s is as defined in A.1.1.1.

5.4.3.2 Receiving

The nominal value of Receiving Loudness Rating (RLR) shall be:

$$RLR = (+\,2 – F_r)\ dB$$

for GATs implementing telephone-band (ITU-T Rec. G.711) speech coding and:

$$RLR = (+\,5 – F_r)\ dB$$

for GATs implementing wide band (ITU-T Rec. G.722) speech coding.

In the above formulae F_r is as defined in A.1.1.2.

The volume control range should span the value of the receiving loudness rating equal to that of the corresponding handset telephone, as well as an RLR value about 10 dB lower.

5.4.4 Echo path loss characteristics

The requirements of ITU-T Recs. P.342 [6] and P.341 [7] respectively apply for the telephone band and wideband operations of GATs intended for connection to the ISDN.

5.4.5 Distortion

The requirements of ITU-T Recs. P.342 [6] and P.341 [7] respectively apply for GATs intended for connection to the ISDN and implementing telephone-band (ITU-T Rec. G.711) or wideband (ITU-T Rec. G.722) speech coding.

5.4.6 Out-of-band requirements

The requirements of ITU-T Recs. P.342 [6] and P.341 [7] respectively apply for GATs intended for connection to the ISDN and implementing telephone-band (ITU-T Rec. G.711) or wideband (ITU-T Rec. G.722) speech coding.

5.4.7 Noise

5.4.7.1 Sending

The requirements of ITU-T Recs. P.342 [6] and P.341 [7] respectively apply for GATs intended for connection to the ISDN and implementing telephone-band (ITU-T Rec. G.711) or wideband (ITU-T Rec. G.722) speech coding.

5.4.7.2 Receiving

Irrespectively from the speech coding implemented, the A-weighted noise produced in the receiving direction at the measurement point by GATs intended for connection to the ISDN shall not exceed –49 dBPa(A)+F_r.

5.4.8 Delay

The requirements of ITU-T Recs. P.342 [6] and P.341 [7] respectively apply for GATs intended for connection to the ISDN and implementing telephone-band (ITU-T Rec. G.711) or wideband (ITU-T Rec. G.722) speech coding.

ANNEX A

Objective measurement methods for hands-free, multiple users GATs

A.1 Introduction

This annex describes methods which may be used to measure the speech transmission performances of hands-free, multiple users, Group Audio Terminals intended for ISDN connection.

Unless stated otherwise, the same test conditions apply as specified in applicable ITU-T Recs. as appropriate (i.e. ITU-T Recs. P.342 [6] and P.341 [7] respectively for GATs implementing telephone-band (ITU-T Rec. G.711) or wideband (ITU-T Rec. G.722) speech coding).

A.1.1 Testing arrangements

The acoustic reference conditions for measurement purposes are free-field within the Reference Sphere. However, for specific designs, it is up to the manufacturer to specify an alternative (reverberant) testing environment. In this case, the specified measurement conditions shall be stated in the test report.

A.1.1.1 Sending

Two different testing conditions shall be used, according to the type of microphone(s) equipping the GAT terminal:

– microphones placed on the table surface in front of one or more speakers:

– microphone(s) placed elsewhere in the room.

Microphones intended for being placed on the table

– The arrangement specified in Figure A.1, placing each microphone in turn at point B. The other microphones shall be removed from the acoustic field of the Artificial Mouth.

Microphones intended for being placed elsewhere in the room

– The acoustic pick-up port of the microphone shall be placed at the centre of the Reference Sphere in an anechoic environment and the Artificial Mouth shall be placed with the lip plane tangential to the Sphere. The reference axis of the Mouth shall coincide with the x-axis of the Sphere, while the microphone shall be oriented as specified by the manufacturer. In case of microphones intended to pick up the sounds coming from multiple directions (either omni-directional or steered devices), the extreme incidence angles on the horizontal plane as stated by the manufacturer shall be tested, as well as an intermediate incidence direction. The testing angles actually used for measurements shall be stated in the test report.

In order to take into account the difference between the reference test positioning and the actual microphone-speaker operating distance (d_s), for which the terminal is adjusted, the following correction factor F_s is defined:

$$F_s\ \text{dB} \quad 20\,\text{Log}\ \frac{d_s}{0.5} \qquad\qquad (d_s \text{ in meters})$$

For sending noise measurements the sound source shall be muted and all microphones shall be placed in the testing environment.

NOTE – The specified methodology is not applicable for testing hands-free terminals equipped with lapel-mounted (lavalier) microphones, the testing method for which is not yet defined.

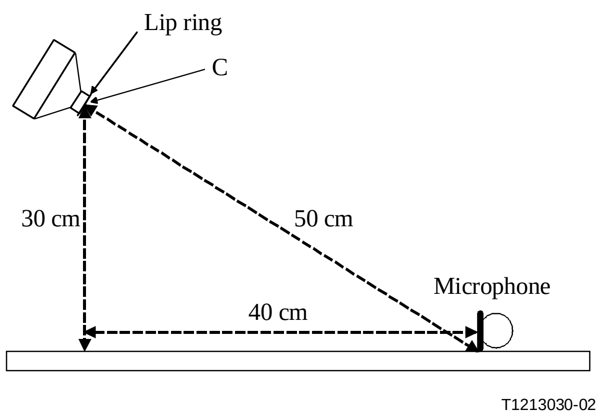

Figure A.1/P.300 – Measurement set-up for table placed microphones

A.1.1.2 Receiving

Each loudspeaker shall be placed in turn at the centre of the Reference Sphere, with its main axis coincident with the x-axis and its outer surface containing the z-axis. The measurement microphone shall be placed on the x-axis, its acoustic centre being at the intersection with the Reference Sphere.

In order to take into account the difference between the reference test positioning and the actual loudspeaker-listener operating distance (d_r) for which the terminal is adjusted, the following correction factor F_r is defined:

$$F_r \text{ dB} \quad 20 \text{ Log } \frac{d_r}{0.5} \qquad (d_r \text{ in metres})$$

A.1.1.3 Terminal coupling loss (TCL)

For TCL and stability measurements the more efficient loudspeaker shall be placed at the centre of the Reference Sphere, with its main axis coincident with the x-axis and its outer surface containing the z-axis. The more efficient microphone shall be placed on the x-axis, its acoustic pick-up port being at the intersection with the Reference Sphere and its maximum efficiency pick-up direction parallel to the x-axis. For surface effect microphones, a suitable horizontal positioning plane shall be used with a minimum surface still guaranteeing the proper operation of the microphone. In any case, the positioning surface shall not enter the reference sphere by more than 100 mm.

In order to take into account the difference between the reference test positioning and the actual loudspeaker-to-microphone operating distance for which the terminal is adjusted, the following correction factor F_{tcl} is defined:

$$F_{tcl} \text{ dB} \quad 20 \text{ Log } \frac{d_{min}}{0.5}$$

d_{min} (in metres) is the minimum loudspeaker-to-microphone distance specified by the manufacturer. If no specification is provided, a default d_{min} = 2 m value shall be assumed and stated in the test report.

A.1.2 Test signals levels

A.1.2.1 Sending

The test signal level shall be (–28.7–F_s) dBPa at the Hands-Free Reference Point (HFRP). The characteristics of the Artificial Mouth shall be according to ITU-T Rec. P.51 [9]. Two different methods can be alternatively used for calibrating the Artificial Mouth.

Method 1: at the HFRP

The input signal from the Artificial Mouth is equalized under free field conditions at the HFRP, such that the spectrum is as specified in A.1 and the total level in the frequency range corresponding to the third octave bands from 100 Hz to 8 kHz is (–28.7–F_s) dBPa. For calculating SLR and response characteristics the reference point is MRP. The sound pressure level at the MRP shall be calculated (by definition) by adding 24 dB to the sound pressure level at the HFRP.

Method 2: at the MRP

The signal generated by the Artificial Mouth is equalized at the MRP under free field conditions to obtain the spectrum specified in A.1, at a level of –4.7 dBPa in the frequency range corresponding to the third octave bands from 100 Hz to 8 kHz. The spectrum at the MRP is then recorded and the level is adjusted to obtain (–28.7–F_s) dBPa at the HFRP. The spectrum recorded at the MRP is used as a reference for calculating SLR and response characteristics.

A.2 Testing of transmission requirements

A.2.1 Sensitivity-frequency response

A.2.1.1 Sending

The measurement shall be carried out for each microphone, which shall be mounted as specified in A.1.1.1. In case of microphones intended to pick up the sounds coming from multiple directions (either omni-directional or steered devices), the extreme incidence angles on the horizontal plane (as stated by the manufacturer) shall be tested, as well as an intermediate incidence direction. If stated by the manufacturer, the nominal direction shall be used as intermediate. The testing angles actually used for measurements shall be stated in the test report.

The noise signal shall be generated by the Artificial Mouth at the level specified in A.1.2.1.

The spectrum of the output signal shall be measured at the output interface of the reference codec.

The sending sensitivity shall be calculated as follows, according to the calibration method used (see A.1.2.1):

Method 1

The sending sensitivity is given by the difference between the electric spectrum and the acoustic spectrum at the MRP:

$$S_{mJ} \quad 20\text{Log } Vs \quad 20\text{Log } P_m, \;\; P_m \quad P_{HRFRP} \quad 24\text{ dB}$$

where:

20 Log Vs: Electric spectrum;

20 Log P_m: Acoustic spectrum at MRP.

Method 2

The sending sensitivity S_{mJ} is given by the following relationship:

$$S_{mJ} \quad 20\text{Log } Vs - 20\text{Log } P_m \quad \text{Corr} - 24\text{ dB}$$

where:

20 Log Vs: Electric spectrum;

20 Log P_m: Acoustic spectrum recorded at MRP;

Corr: Correction factor (20 Log P_{MRP}/P_{HFRP}) of the Artificial Mouth.

A.2.1.2 Receiving

Each loudspeaker shall be placed in turn in the test room as specified in A.1.1.2.

The noise signal generator shall be connected to the input of the reference codec.

The sensitivity at each 1/3 octave band shall be calculated by subtracting the spectrum of the electric signal from the acoustic spectrum at the measurement point.

The measurement shall be repeated at the minimum and maximum position of the (manual) volume control, changing the input level accordingly. In case of devices not provided with manual volume control, the measurement shall be repeated for excitation levels of –30 dBm0 and –15 dBm0.

A.2.2 Loudness rating

A.2.2.1 Sending Loudness Rating

The sending sensitivity shall be measured for each of the fourteen 1/3rd octave bands given in Table 1/P.79 [10], bands 4 to 17 (200 Hz-4 kHz). The sensitivity is expressed in terms of dB(V/Pa) and the SLR shall be calculated according to Formula A-23b/P.79, over bands 4 to 17 and using the sending weighting factors from Table A.2/P.79, adjusted by subtracting 0.3 dB from each value, using the sending sensitivity response of A.2.1.1.

The measurement shall be carried out for each microphone, which shall be mounted as specified in A.1.1.1. In case of microphones intended to pick up the sounds coming from multiple directions (either omni-directional or steered devices), the extreme incidence angles on the horizontal plane (as stated by the manufacturer) shall be tested, as well as an intermediate incidence direction. If stated by the manufacturer, the nominal direction shall be used as intermediate. The testing angles actually used for measurements shall be stated in the test report.

A.2.2.2 Receiving Loudness Rating

The receiving sensitivity shall be measured for each of the fourteen 1/3rd octave bands given in Table 1/P.79 [10], bands 4 to 17 (200 Hz-4 kHz). The sensitivity is expressed in terms of dB(Pa/V) and the RLR shall be calculated according to Formula A-23c/P.79, over bands 4 to 17 and using the receiving weighting factors from Table A.2/P.79, adjusted by subtracting 0.3 dB from each value, using the receiving sensitivity response of A.2.1.2.

The receiving sensitivity shall not be corrected by the L_e factor. The calculated RLR shall be corrected by subtracting 14 dB, according to ITU-T Rec. P.340 [5].

A.2.3 Terminal Coupling Loss

The measurements shall be carried out by coupling, as specified in A.1.1.3, the most efficient loudspeaker with the most efficient microphone.

A.2.3.1 Weighted Terminal Coupling Loss

The test signal shall be a pink noise, with a level of $(-15-F_{tcl})$ dBm0.

The attenuation from digital input to digital output shall be measured at the 1/3rd octave frequencies given by the R10 series of preferred numbers in ISO 3 [11] for frequencies from 100 Hz to 8000 Hz.

The TCLw shall be calculated according to the method in B.4/G.122 [12] (trapezoidal rule), on the frequency band from 100 Hz to 8 kHz. The value calculated according to ITU-T Rec. G.122 [12] ($TCLw_{calculated}$) shall be corrected by adding the correction factor F_{tcl} and by considering the total sound power produced by all the loudspeakers:

$$TCL_w = TCL_{w_{calculated}} + F_{tcl} + 10\mathrm{Log}\,\frac{n+1}{2}$$

where n is the total number of loudspeakers equipping the terminal under test.

NOTE – The formula above assumes that only one microphone is active at each time and that the sound pressure levels of signals picked up from the more distant loudspeakers is 3 dB less than the signal level generated by the better coupled loudspeaker. It is also assumed that a power summation law applies to the signals picked up by the different loudspeakers. This is not absolutely true in principle, but some un-correlation of the sound signals generated by the different loudspeakers can be assumed to result from the different transmission path lengths and from the reverberation effects occurring in actual use.

A.2.3.2 Stability loss

The test signal shall be sinusoidal, with a level of $(–15–F_{tcl})$ dBm0.

The attenuation from digital input to digital output shall be measured at 1/24th octave intervals for frequencies from 100 Hz to 8 kHz. The actually measured values shall be corrected by adding F_{tcl} and by subtracting 10 Log [(n+1)/2] (as for the TCLw measurement).

APPENDIX I

Practical *in-situ* audio alignment procedure for multiple-users hands-free GATs

I.1 Introduction

The hands-free function for a group of users is normally provided with audio arrangements which widely interact with the installation environment and a guide is provided here for *in-situ* aligning the electroacoustic sensitivities and checking the correct transduction across the whole frequency band.

I.2 Audio alignment and sensitivity-frequency characteristics

Audio alignment is an operation carried out when the terminal is installed, and any time the acoustic configuration of GATs is changed. It is intended to guarantee the proper setting of sending and receiving level adjustments, in order to prevent the channel overloading while still guaranteeing adequate signal to noise and a comfortable receiving audio level.

The following equipment should be used for performing the audio alignment of GAT terminals:

– Artificial Mouth (complying with ITU-T Rec. P.51 [9]);
– signal generator (digital or analogue);
– sound level meter (complying with IEC 60651 [13]);
– electric level meter.

The acoustic test signal generated by the Artificial Mouth should consist of speech-shaped noise (ITU-T Rec. P.50 [14]), at a level of –4.7 dBPa at the MRP.

The signal generator is used for performing the receiving alignment and can either generate an encoded digital path at the digital interface of the GAT, or an analogue signal to be applied at a suitable analogue test point. In the latter case, a reference codec could be necessary for digital GATs if no specific interface for alignment purposes is available within the terminal. The signal for receiving alignment should be a speech-shaped noise (ITU-T Rec. P.50 [14]) at a level of –20 dBm0. If necessary, the noise can be modulated ON-OFF. In this case, the signal level is defined as the level generated during the ON periods of the signal.

NOTE – Terminal manufacturers are encouraged to implement an analogue 0 dBr interface for alignment purposes in their equipment.

If no specific alignment interface is provided, the electric level meter for sending alignment should either directly process the digital pattern generated by the terminal, or be used in conjunction with a Reference Coder.

I.2.1 Sending

I.2.1.1 Sensitivity adjustment(s)

The Artificial Mouth should be placed over the border of the conference table, as shown in Figure A.1, in correspondence of each conferee position. The conference microphones should be positioned on the conference table (or in the conference room) as in normal use.

If a sensitivity adjustment is provided for each microphone, it should be regulated in turn in order to achieve an output level of –20 dBm0 when the Mouth is placed in correspondence with the associated microphone. Otherwise, if only one sensitivity control is provided, it should be regulated in order that the average reading, obtained by placing the Mouth in correspondence of each conferee position in turn, is equal to –20 dBm0.

I.2.1.2 *In-situ* frequency response

The *in-situ* measurement of the sending frequency response is defined as the difference between the octave spectra of the output signal and of the acoustic excitation signal at the MRP. Considering that the measurements are *in-situ*, the octave band analysis shall be restricted to the bandwidth from 125 Hz to 4 kHz (centre frequencies). It is recommended that the sum of the absolute differences between the measured values and their average is less than 8 dB.

I.2.2 Receiving

I.2.2.1 Sensitivity adjustment

The speech-shaped noise is fed to the input of the terminal at a level of –20 dBm0. With the manual volume control (if any) at its maximum position, the receiving gain should be adjusted in order to achieve a sound pressure level of at least 54 dBA at each conferee position. The measurements should be made by placing the measurement microphone at point C (Figure A.1) corresponding to each conferee position, and in the absence of the Artificial Mouth.

NOTE – Practical experience has shown that higher listening levels, up to 65 dBA, may be desirable. The maximum settable gain being constrained by the terminal stability limit.

I.2.2.2 *In-situ* frequency response

The *in-situ* measurement of the receiving frequency response is defined as the difference between the octave spectra of the output signal, and of the input excitation signal at the terminal interface. Considering that measurements are *in-situ*, the octave band analysis shall be restricted to the bandwidth from 125 Hz to 4 kHz (centre frequencies). It is recommended that the sum of the absolute differences between the measured values and their average is less than 12 dB.

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